SCALABLE VOICE OVER IP SYSTEM PROVIDING INDEPENDENT CALL BRIDGING FOR OUTBOUND CALLS INITIATED BY USER INTERFACE APPLICATIONS

ABSTRACT OF THE DISCLOSURE

An IP telephony gateway and a user interface resource enable a subscriber to place an outgoing call according to the voice over IP (H.323) protocol to a destination party from a user interface session of an intelligent dial tone service such as voice activated dialing, and resume the user interface session upon completion of the outgoing call with the destination party. The IP telephony gateway establishes a user interface session for the subscriber with the user interface resource across a first Real Time Protocol (RTP) data stream. The user interface resource initiates a second RTP data stream to a destination party in response to reception of a prescribed command from the subscriber. Although an RTP bridge connecting the first and second RTP data streams can be maintained by the user interface resource, the user interface resource may also use the Empty Capability Set feature in the H.323 standard to cause the IP telephony gateway to close the first and second RTP data streams to the user interface resource. The user interface resource then issues Non-Empty Capability Set messages to the IP telephony gateway for the first and second RTP data streams, causing the IP telephony gateway to internally bridge the first and second RTP data streams. The user interface resource monitors connections between the subscriber and the destination party, and upon detecting a disconnect by the destination party causes the IP telephony gateway to resume the user interface session, by repeating the sequence of sending Empty Capability Set and Non-Empty Capability Set messages to the IP telephony gateway to break down the bridge and re-establish the connection between the subscriber and the user interface resource.